REMARKS

The principle difference between the present invention and the prior art is that the same stored test voice information is stored at every node or terminal where a speech quality measure might be required and that voice and test information are carried on the same call. The reason that this is so important is as follows:

by having stored data test voice information at each node or terminal all users have an exact copy of something with which to make a comparison;

introduction of the test voice information with a real voice payload ensures that on transmission to another node terminal the test voice information and the voice suffer the same effects which means that the speech quality measure is useful;

on reception at the other node or terminal a comparison between the stored test voice information at that node or terminal and the test voice information which has been transmitted gives an exact measure of the impact of the network on speech quality.

The prior art either in the form of Lewis or any combination of Lewis, Tschudin and Petitcolas does not teach the present invention. No reference teaches the critical step of providing each node or terminal with the same stored test voice information and sending test data and voice data embedded together on the same call.

Lewis teaches a system which sends voice and test data on different calls. This means it is likely that the voice and test data do not travel the same way and as such do not suffer the same impacts from the network. There is no stored test voice information at each node/terminal in the network. As such Lewis cannot give an accurate measure of the speech quality of a specific call in the network and suffers from many of the drawbacks identified in the introduction of the present application.

Tschudin describes a method to provide confidentially without encryption, by introducing "chaff" (fake packets) into the true packet sequence (the "wheat"). These packets are introduced based on a pseudo random sequence, with this sequence known only to the sender and receiver. These packets are not tagged in a real sense, but identifiable via the pseudo random sequence (this could only be considered virtual tagging).

Thus, Tschudin is not describing a technique where test information is added to a packet stream. Rather he describes a system where the packet stream itself is hidden in a new packet stream. Accordingly there is no teaching of the novel feature of the present patent application.

Petitcolas, in general, also teaches information hiding, rather than the embedding of test data in the true signal. In the specific section quoted (pg 1067 section D) there is no mention of the test information.

Accordingly the applicants believe the application is now in order for issue, and such action is solicited.

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Individual packets for the voice call between A and B may take different routes between A and B as explained above. This means that the packets comprising test voice information may follow different routes from the packets comprising the voice information for the ongoing voice call. However, because the packets are all sent as part of the same voice call (for example, as part of the same RTP session), the test voice information packets experience approximately the same effects from transmission through the network as do the real voice information packets. This provides the advantage that an improved assessment of the amount of degradation experienced by the voice call is obtained. Previous methods that have used dummy test packets (which contain no test or real speech information) to measure percentage packet loss provid[e]ing a different type of assessment. Other types of previous method have used dedicated calls for test speech to enable end to end testing. In that case the test speech does not enable an accurate assessment of a particular voice call as in the present invention. In addition, many dedicated voice terminals can handle only one call at a time, so a separate call for test speech is not possible.

Claims

1. (Amended) A method of measuring the speech quality of a voice call between a first node and a second node in a packet-based communications network, each of the first and second nodes comprising the same stored test voice information, the method comprising the steps of, at the first node:

(i) receiving packets for the voice call and adding at least part of the stored test voice information to at least some of the packets;

(ii) forwarding the packets to the second node;

(ii) at the second node, accessing the stored test voice information [stored] at the second node and comparing it with the test voice information received in the packets using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.

11. (Amended) A signal for a voice call provided over a packet-based communications network, said signal comprising a plurality of packets at least some of which comprise test voice information for comparison at a node with stored test voice information which is the same as the test voice information.

14. (Amended) A packet-based communications network node arranged to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party wherein the caller and the called party each have stored test voice information, said node comprising:

(i) an input arranged to receive packets for the voice call; and

(ii) a processor arranged to add test voice information to one or more of the packets;

(iii) an output arranged to forward the packets towards the called party for comparison of the test voice information with the stored test voice information of the called party to provide a measure of said speech quality.

17. (Amended) A communications network comprising a first [node as claimed in claim 14] packet-based communications network node arranged to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party wherein the caller and the called party each have stored test voice information, said node comprising:

(i) a first input arranged to receive packets for the voice call; and

(ii) a first processor arranged to add test voice information to one or more of the packets;

(iii) a first output arranged to forward the packets towards the called party for comparison of the test voice information with the stored test voice information of the called party to provide a measure of said speech quality;

and a second [node as claimed in claim 16.] packet-based communications network node arranged to measure speech quality for a call which is ongoing between a caller and a called party, said node comprising:

(i) <u>a second input arranged to receive packets as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;</u>

(ii) <u>stored test voice information;</u>

(iii) <u>a second processor arranged to compare the received test voice information using a speech quality assessment algorithm in order to obtain a measure of speech quality assessment algorithm in order to obtain the measure of speech quality for the voice call.</u>

18. (Amended) A method of measuring speech quality for a call which is ongoing, said method comprising, at a node in a packet based communications network:

(i) receiving packets as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) accessing stored test voice information <u>at the node</u>;

(iii) comparing the received test voice information and the accessed stored test voice information using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.

19. (Amended) A method of enabling speech quality to be measured for a voice call which is ongoing between a caller and a called party said method comprising, at a node in a packet based communications network:

(i) receiving packets for the voice call;

(ii) adding test voice information to one or more of the packets; and

(iii) forwarding the packets towards the called party;

(iv) at the called party node extracting the received test voice information and comparing it with stored test voice information at said called party node to provide a measure of said speech quality.

20. (Amended) A computer program for controlling a packet-based communications network node in order to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party said computer program being arranged to control the node such that:

(i) packets for the voice call are received;

(ii) test voice information is added to one or more of the packets; and

(iii) the packets are forwarded towards the called party;

(iv) at the called party node the received test voice information is compared with stored test voice information at said called party node to provide a measure of said speech quality.

21. (Amended) A computer program arranged to control a packet-based communications network node in order to measure speech quality for a call which is ongoing between a caller and a called party, said computer program being arranged to control the node such that:

(i) packets are received as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) stored test voice information [stored] at the node is accessed; and

(iii) the received test voice information and the stored test voice information are compared using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.